Rec'd PCT/PTO 12 SEP 2005

10/519590

Wireless Audio Signal Transmission Method for a Three-Dimensional Sound System

The invention relates to a wireless audio signal transmission method for a three-dimensional sound system. In the home setting, modern audio reproduction systems are increasingly intended to provide multichannel sound reproduction based on the Dolby digital standard, the DTS standard (= "Digital Theater System"), or other three-dimensional sound method, in combination with a television receiver for digital reception or with a DVD player (= "digital versatile disc"). With these systems, the audio signals are transmitted to up to six different speaker locations. In the home setting, however, the required installation of signal lines is often a problem. For this reason, there is often a desire to have wireless transmission which in addition enables playback devices and speakers in different rooms to be interconnected.

Solutions currently already on the market are based on transmission links using frequency modulation. However, the quality of this analog transmission for speakers or headphones usually does not meet more demanding requirements. In addition, analog transmission is susceptible to interference, is not secure against being intercepted, and is inefficient in utilizing the available bandwidth. In the home setting, disturbed reception conditions are also to be expected due to reflections and shadowing. A first step for improvement is to replace the analog signal transmission by the transmission of data which have been generated by the prior sampling and digitization of the analog signals. An example of wireless audio signal transmission is presented in the applicant's own patent application EP 0 082 905 A1. Using an infrared transmission device, digitized audio signals are transmitted by a transmitting device, for example, a television receiver, to "active speaker boxes" which can be set up at any location within the room. The need for the inconvenient signal lines is thus eliminated, while only connections to the standard AC power supply are required to provide power – normally accomplished without difficulty. Unfortunately, this system is suitable only for stereo signals and not applicable for multichannel sound system techniques.

The goal of the invention is therefore to provide, for a three-dimensional multichannel sound system, a wireless audio signal transmission method and associated transmitter and receiving devices, which avoid the above-described disadvantages without increasing the cost by an unreasonable amount, and wherein the audio signal transmission method is also suitable for controlling headphones, including for a stereo operating mode.

Accomplishment of the goal is implemented based on the features of Claim 1 by first digitizing the relevant audio data for one or more audio signal transmitting devices, then transmitting them by a digital modulation method as symbols. The number of required high-frequency channels is governed here by the bandwidth specified by legislators for each channel, and on the total bandwidth of the frequency range used. The already relatively interference-proof transmission using symbols is further improved by employing a diversity method. Modified transmitter and receiving devices are placed under patent protection in dependent Claims 9 and 12.

Interference caused by multipath reception and shadowing is avoided by an appropriate diversity method. The propagation of HF and UHF signals within spaces is characterized chiefly by a multiplicity of mutually independent propagation paths from the transmitter to the receiver. In addition to a more or less strongly attenuated direct path, multiple indirect paths arise, depending on whether or not obstacles are present. Since the path lengths differ, the individual signals arrive at different phase positions. When the phase offset is exactly 0°, 360°, or a multiple thereof, this is known as constructive interference. If, on the other hand, this offset is 180°, or 180° plus a multiple of 360°, then this is destructive interference. If the two signals are equally strong, then the result in this case is total cancellation of the two signals. This effect is, of course, dependent on frequency since the phase shift over a fixed path length is a function of frequency. Field strength measurements between a transmitter and a receiver for which a movement occurred in an indoor space over a 15 m path having reflections and obstacles showed field strength drops of up to 30 dB at a frequency of 864 MHz, where, it must be noted, the direct propagation path was attenuated by an obstacle.

In today's FM wireless speakers, one tries to avoid this situation through the careful placement of the receiver. Since, however, people must also be taken into account as obstacles or reflectors, their movement results in a constant change in propagation conditions. This is of course especially true if the receiver is portable, as in the case with battery-powered headphones which are also designed to have a wireless connection to the transmitting device and are thus equipped with a corresponding receiving device.

The simplest solution would be to increase the transmission power. For legal reasons, this is not possible with the available frequencies. Since the interference effects are a function of location and path, the obvious solution is to implement two or more mutually independent transmission paths using a

diversity method. The frequency dependence of the interference phenomena can be exploited by transmitting on two different frequencies simultaneously, then selecting the better signal on the receiver side. This solution is not economical in terms of frequency; it is thus in conflict with the goals of the transmission concept. A much more common approach is receiver diversity. In order to maintain the independent paths needed for propagation, two receiving antennas are set up at a distance of at least $\lambda/4$ from each other. Now either the relatively stronger antenna signal is selected by the receiver, or the two signals are combined. In order to avoid drop-outs during switching, this approach requires, however, that at least two receivers in complete form up to recovery of the channel-coded data be provided at each receiving site.

The following discussion explains the invention and advantageous embodiments in more detail based on the figures of the drawing:

Figure 1 is a schematic view illustrating the known transmitter and receiver diversity methods;

Figure 2 is a schematic view illustrating the known transmitter diversity method;

Figure 3 is a schematic view illustrating the known receiver diversity method;

Figure 4 is a schematic view illustrating a transmitter diversity method used for the invention with identical transmission frequencies to transmit data;

Figure 5 shows the transmission scheme with the associated "space-time-block code";

Figure 6 is a schematic view illustrating a transmitter according to the invention in the form of a block diagram;

Figure 7 is a schematic view illustrating a receiver according to the invention in the form of a block diagram;

Figure 8 shows two different data formats according to the invention.

The first advantage provided by digitization of the audio signals to be transmitted is the higher level of immunity against interference due to quantization which can then be further enhanced through the addition of check bits or other error-detection or error-correction methods. The second advantage is the fact that, on the data level, there are a sufficient number of known methods for data reduction that specifically involve the redundant properties of the respective signals to reduce the amount of data without any loss in quality.

Unfortunately, the use of a diversity method increases the number of channels to be transmitted. When using diversity methods, normally one transmitter and one receiver are required for each transmission channel – see Figure 1. If in the simplest case each audio channel is designed in duplicate form, then the resulting requirement for six speaker sites is 12 high-frequency channels, and an equal

number of transmitters, receivers, and antennas. This approach would make cost-effective implementation impossible. Figure 1 shows an example of such a diversity method with two channels, in which a signal source Q is connected to a reproduction device LB, for example, a speaker box, through two transmitters S1, S2 with two antennas AS1, AS2, and two receivers E1, E2 with two antennas AE1, AE2, where the signals transmitted through transmitting antennas AS1, AS2 have different transmission frequencies f1, f2. Evaluation of the received signals and generation of the actual audio signals is implemented in an attached electronics system E3. Diversity is achieved based on the frequency-dependent propagation conditions for the two transmission frequencies f1, f2, since the phase positions due to reflections and obstacles vary, and generally given different frequencies an attenuation or even cancellation occurs, with the result that one of the received signals always has sufficient field strength. Additional improvements are possible based on not only exploiting the frequency diversity but also by making the spacing between the transmitting antennas or spacing between the receiving antennas as large as possible, or by making the polarity and emission direction or reception direction different relative to each other. These measures can be carried out singly or in combination. A further improvement can be achieved not only by having the two receivers E1, E2 each detect one of the two different transmission frequencies, but also by designing them to be as broadband as possible such that both frequencies are received. Separation of the frequencies and their contents is then effected internally by filtering means. The number of transmission paths is then doubled so that the feared cancellations are even less likely.

A simplification of this complex approach is provided by one-sided diversity methods which have separate transmission channels or receiving channels either only on the transmitter side, see Figure 2, or only on the receiver side, see Figure 3, opposite which channels is a single receiver E4 or transmitter S3. In the transmitter diversity method of Figure 2, the signals to be transmitted are transmitted by two transmitters S1 and S2, and two antennas AS1 and AS2, on two different frequencies f1, f2. On the receiver side, the two signals passing along different propagation paths are superimposed on each other and are detected by a single antenna AE with an associated receiver E4. The transit time differences due to the frequency diversity and space diversity generally prevent any simultaneous total cancellation of the two frequencies f1, f2. In receiver E4, either the signal content of the two frequencies f1, f2 is heterodyned, or that frequency is selected which at that instant has the higher field strength. A special case from Figure 2 – not illustrated however – employs the same transmitting antenna for both frequencies f1, f2. In this case, only frequency diversity remains.

In the receiver diversity method of Figure 3, only a single transmitter S3 is present which transmits the signal at the transmission frequency f through its antenna AS. On the receiver side, this signal is received by two separate antennas AE1, AE2 and associated receivers E1, E2 to which, as in Figure 1, a common electronics system E3 is attached which ultimately feeds the reproduction device LB. This method involves space diversity, although directional diversity or polarity diversity can be added through receiving antennas AE1, AE2. The two signals from receivers E1, E2 are either heterodyned in the attached electronics system E3, or this system has a selection circuit which further-processes only the antenna signal that has the higher field strength.

Receiver diversity is commonly employed, for example, in professional settings for portable microphones since this situation does not allow for multiple transmitting antennas. The frequency-modulated signal from the microphone transmitter is received here by the associated receiver which is coupled to two extendable antennas, each of which is attached to a high-frequency receiver. While the diversity method is not optimal here due to the relatively close spacing of the receiving antennas, the cost/complexity of the electronics involving sensitive receivers, the further relaying and processing of the signals are not, of course, of importance; if necessary, one simply utilizes an additional receiver.

For applications in the home setting, multiple antennas in speaker boxes are simply out of the question for aesthetic reasons. Diversity methods based on Figure 1 and Figure 3 are thus eliminated from consideration. Fortunately, there exists a modified transmitter diversity method which is a refinement of the approach in Figure 2; however, this is capable only of transmitting data sequences. The added expense of this method in terms of equipment is essentially only on the transmitter side, not the receiver side. Figure 4 shows the relevant transmitter and receiver diagram. Figure 5 is a schematic view in the form of a table showing how the transmission of two different data sequences is implemented using the same transmission frequencies, as in the known space-time block code method. The principles of this method are described in detail for different variants, for example, in "IEEE SIGNAL PROCESSING MAGAZINE," May 2000, pages 76 to 91, in the article "Increasing Data Rate over Wireless Channels" by Ayman F. Naguib, Nambi Seshadri, and A. R. Calderbank. In order to be able to use this method to control high-end audio reproduction devices LB, the source Q must supply data as audio signals, or, in the case of analog signals, digitization must occur in source Q or in an attached encoder CS.

In the transmitting device S40, the data stream D_o to be transmitted is processed within encoder CS, as shown in Figure 5, as a first and second data stream D_1, D_2, supplied to transmitter stage S4 with its high-frequency transmitters S5, S6, then transmitted through two spatially separated antennas AS1, AS2 as quadrature-modulated signals, but in the same frequency band f despite the different contents. On the receiver side in the receiving device E50, a single high-frequency receiving device AE, E5 with appropriately adapted decoder CE is sufficient to recover the original data sequence D_o from the heterodyned signals r or a data sequence Dr generated therefrom. This data sequence is then available for further processing and reproduction in audio reproduction device LB. The fact that this diversity method is applicable – disregarding the expense factor – only for the transmission of data is not a disadvantage since, as is well known, the transmission of data is less susceptible to interference than is the transmission of audio signals, and given appropriate encoding requires less channel width. When the audio signal is converted to a data stream by digitization, it is also possible to apply known techniques of data compression.

A further simplification is achieved by data compression on the transmitter side. The available high-frequency channels are relatively narrow-band and have a maximum channel width of, for example, 300 kHz. Using data compression, however, it is nevertheless possible to transmit data from two or more audio channels on one high-frequency channel. The data compression here exploits the redundancy in the audio signals, the right information and left information of symmetrical speaker locations being especially well-suited for this type of compression. For the purpose of actual transmission, the data stream is then converted into symbols which are transmitted by the high-frequency carrier.

The digital transmission provided, that is, the transmission of symbols, requires on the receiver side an evaluation of the received signal at predefined times at which the transmitted signal occupies a defined state in the quadrature signal plane. In order to determine this state which corresponds to the transmitted symbol – but more or less disturbed due to the transmission – the received signal is sampled and digitized, at least at defined times. The elimination of interference, subsequent conversion, and decoding are then also implemented purely digitally. In zero-IF or low-IF receivers in which the two

quadrature components are converted directly to the baseband or a low frequency position where they are digitized, especially cost-effective receiving concepts can be provided which can be accommodated within a single IC for each receiver, and which manage without significant external circuit elements. Since after frequency conversion the decoding and subsequent signal processing are implemented in one digital signal processor, any inaccuracies in the analog component of the circuit, such as phase errors or amplitude errors, can be corrected in this processor since asymmetries and inaccuracies as separate error sources are not possible in the digital processing component.

In terms of selection of the transmission band, a number of high-frequency bands are available. It is advantageous to take a transmission band which is available for transmissions of this type. The approved frequency range between 433.020 MHz and 434.790 MHz, also known as the "ISM band," is less well suited since in this range there is no protection from other users or from the priority-status transmissions of amateur radio. Not only would one's own alarm system or the wirelessly controlled central locking system of the neighbor's car interfere: the FM signal can be intercepted by anyone. As of now, the 863 MHz to 865 MHz frequency band reserved for audio transmission has found only reluctant acceptance, presumably because the 10 mW approved radiated power (ERP) is relatively low for operation not subject to individual certification. Within close range, the use of this frequency band for the wireless control of audio reproduction devices would be quite suitable as long as the transmitting and receiving antennas are within sight of each other. If this is not the case, degradations in reception result. As already mentioned, the signal is not only subject to attenuation but also to multiple reflections. Whenever two of these signal components now arrive at the receiver in phase opposition but with approximately the same intensity, they cancel each other completely. This is what's known as interference. In the extreme case, an almost complete loss of reception may result.

A frequency band around 40 MHz can be eliminated from consideration due to the narrow bandwidth. Strong interference must be expected in the segment around 432 MHz in the 70-cm amateur band. Frequencies in the GHz range can be eliminated based on the higher component costs and increasingly unfavorable propagation conditions. In addition, the lowest portion of this range around 2450 MHz is already home to a number of services and users such as Bluetooth, wireless data links, and microwave ovens. What remains is thus the range around 864 MHz, especially considering, for example, that this range is specifically intended for wireless audio applications in streaming mode (duty cycle = 1), that is, the high-frequency carrier in each channel can be in action continuously. Due to the limited bandwidth of only 2 MHz for this entire frequency band, the audio data have to be compressed. To provide simultaneous video reproduction, lip-synchronicity is required, with the result that the maximum

allowable delay between video and sound is 20 ms. In light of the chosen compression method plus the obvious demand for the highest possible fidelity of reproduction, this requirement must be adhered to. Appropriate compression methods by which computationally to compress the 16-bit or 24-bit audio data to 6 bits per sampling value are known – see, for example, ADPCM (= adaptive differential pulse code modulation) or other methods in "K. D. Kammeyer," *Nachrichtenübertragung* [Information Transmission], B. G. Teubner Stuttgart, 2nd edition 1996, pages 124 through 137, Chapter 4.3 "Differential Pulse Code Modulation." A stereo signal sampled at 48 kHz would thus yield a data rate of 576 kB/s. Higher-level compression methods such as MP3 which would enable a stronger compression are not suitable since their delay is too large, and a transmitter-side preliminary delay of the video information in the home setting is too complex.

The 16-QAM method is advantageously selected as the digital modulation approach to transmit the symbols. This represents a favorable compromise between transmission capacity and implementability. Extensive system analyses show that a 3/4 trellis coding of the modulation provides for sufficient error protection. The gross data rate for the stereo signal is thus around 768 kB/s. Synchronization and control of the spatially distributed audio reproduction devices require a small number of additional data to be transmitted such that the final data rate is approximately 840 kB/s. The thus obtained symbol rate of 210 kS/s can be accommodated with a roll-off factor of 19% within a 250-kHz-wide channel. As a result, eight HF carriers, each with two audio channels, are available within the 2-MHz-wide segment between 863 MHz and 865 MHz.

A fully expanded system having six-channel sound does require three of the eight HF channels, with the result that only two of these systems can be operated in parallel within a house without interfering with each other. Experience shows, however, that often the center and sub-loudspeaker are connected directly by wire to the playback device, with the result that only two HF channels are needed. In addition, the system provides for dynamic assignment of the channels, with the result that only one carrier need be used for one stereo signal, even when more than two speakers are operated. The fundamental consideration – that two antennas set up to be sufficiently separated from each other on at least one side of the transmission path, with a single antenna on the opposite side, form two mutually independent transmission links – is also valid in the case in which the two antennas are located on the transmitter side. Of course here, where a backward channel is lacking, the transmitter cannot choose

between the two antennas since it does not have any information about the respective reception conditions. As a result, a way must be found to transmit the useful signal twice so as to obtain the diversity gain, without simultaneously causing a mutual degradation of the two signals. An obvious solution here is the above-mentioned space time coding method, whose space-time block codes (STBC) or space time trellis codes (STTC) meet this requirement.

The table of Figure 5 is a schematic view showing the STBC coding and transmission of a data sequence D_o with data A, B, C, D. The first line "clock" indicates the successive clock times T_1, T_2, T_3, T_4 for the original data sequence D_o and transmission of the symbols. The original data sequence D_o with data A, B, C, D is in the second line. The third and fourth lines show a first data sequence D_1 obtained by conversion with the data A, -B*, C, -D*, D2, and a second data sequence D_2 with the data B, A*, D, C*. The third and fourth lines represent the symbol sequences which are transmitted using quadrature signals by the two antennas AS1 and AS2. The asterisk * here indicates the complex conjugate data value. The fifth line defines the even and odd times "even" and "odd" for the times T_1 through T_4. Finally, the sixth line shows the combination of symbols A, B, and C, D to form a first or second symbol pair Sy1, Sy2. For the sake of completeness, it must be mentioned that the data sequences D_1, D_2 may also be combined differently, for example, D_1 with A, B*, C, D*, and D_2 with -B, A*, -D, A*, or other combinations. It must simply be ensured that symbols A, B, C, D are coded differently in the two data sequences and that the appropriate equations are available on the receiving side.

In the first step during time T_1, the two successive symbols A, B are transmitted in parallel. Antenna AS1 transmits symbol A, and antenna [*AS2*][1] transmits symbol B. For the purposes of differentiation, in the referenced literature the two successive symbols A, B are identified as a symbol pair, first symbol A being defined as the even symbol, and second symbol B being identified as the odd symbol. Subsequently, transposition and transformation of the two initially transmitted symbols A, B takes place, with the result that in the second step during time T_2 at antenna AS1 the symbol B is transmitted in the form of the complex conjugate and negated as -B*, while the other symbol A is transmitted in the form of the complex conjugate as A*. After two steps T_1, T_2, a symbol pair A, B, the first symbol pair Sy1, is thus transmitted. During the third and fourth times T_3, T_4, the second symbol pair Sy2 with symbols C, D is transmitted in identical fashion. Each symbol is thus transmitted twice. Since,

[1] "AS2" added by translator.

however, there is also a parallel transmission through both antennas AS1, AS2, the data rate for the data sequence D_r on the receiver side is identical to the original data rate of data sequence D_o.

On the receiver side, the symbols A, B, or C, D received at the same frequency and superimposed must now be separated. Mathematically, this corresponds to the solution of a linear equation system with two unknowns A and B:

$r_{even} = h1 \cdot A + h2 \cdot B$		equation (1)
$r_{odd} = h2 \cdot A^* + h1 \cdot (-B^*)$	by transformation produces	equation (2)
$r_{odd}^* = h2^* \cdot A - h1^* \cdot B$		equation (3)

Here h1 denotes the transfer function from first antenna AS1 to receive antenna AE, while h2 denotes the transfer function from second antenna AS2 to receive antenna AE. The received signal value at time "even" is r_{even} and is composed of components A and B, and the two transfer functions h1 and h2. The received signal value r_{odd} at time "odd" is composed of the components h1, h2, A* and -B*. As long as transfer functions h1 and h2 are known, the equations (1) and (2) represent a linear system from which A and B can be determined. If the complex conjugate form corresponding to equation (3) is generated from both sides of equation (2), then the symbols A, B are identical with the symbols of equation (1).

The transfer functions h1, h2 are initially unknown. However, they represent, as it were, a steady state since the spatial conditions relative to the data rate only change relatively slowly. In addition, one can start with the useful assumption that both transfer functions are initially equal, then seek the optimum setting by means of a control action on the receiver side. To this end, the received signals on the receiver side are multiplied by an inverse transfer function in a linear combination h^{-1} (see Figure 7) which is initially present as an estimated value, then is adapted by an adaptive algorithm to the actual transfer functions of the two transmitting antennas AS1, AS2. The transfer functions h1 and h2, along with their associated inverse transfer functions h_1^{-1} and h_2^{-2}[2] in the linear combination h^{-1} together form a linear frequency response. Based on the linear combination h^{-1}, the symbols A', B' received after the transfer are translated into the quadrature signal plane such that a symbol decision element ET can determine the associated decided symbols A", B" from these values. If as a result of transfer changes

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[2] Translator's note: exponent corrected from context.

in received symbols A', B' deviations occur relative to the inverse transfer functions h_1^{-1} , h_2^{-2} in the linear combination h^{-1}, then these deviations are detected essentially as differences by an equation system in an arithmetic unit RE. These difference values are then smoothed by a control loop filter Fr and supplied as correction values of linear combination h^{-1}.

Figure 6 shows the essential functional units of an embodiment of a transmitting device S40 according to the invention in the form of a block diagram. A signal source Q supplies an analog audio signal to an analog-to-digital converter AD, the output of which supplies a data stream D_o with the symbol rate determined by the digitization clock t_s. The digitization clock here advantageously corresponds to the symbol clock t_s generated in a symbol clock generator T_s, or a multiple thereof. The two different data streams D_1 and D_2 are generated from data stream D_o in a transmission coding device CS, which data streams contain the individual symbol pairs A, B; C, D, but with the respective different coding in the quadrature signal plane as shown in Figure 5. In a high-frequency stage S4, the two data sequences D_1, D_2 are transferred to the desired high-frequency band by means of the sine and cosine components of a quadrature carrier tr coming from a high-frequency oscillator Os1, then transmitted separately through antennas AS1, AS2. For the sake of clarity, the required pulse form filter, as well as the filter devices to avoid interference and alias signals, are not shown in Figure 6.

Figure 7 is a schematic view in the form of a block diagram illustrating an embodiment of a receiving device E50 according to the invention. Circuit unit E5 is a heterodyne receiver which uses a high-frequency mixer M3 to convert the high-frequency signal received through antenna AE from the high-frequency channel f to an intermediate frequency position which lies approximately in a frequency range of 1 to 2 MHz. The carrier for mixer M3 is a high-frequency signal HF from a local oscillator Os2. After mixer M3, a bandpass F1 filters out the desired frequency band and feeds the filtered signal to an analog-to-digital converter ADE for digitization. The conversion to an intermediate frequency has the advantage that only a single analog-to-digital converter ADE is required. In the case of zero-IF conversion or low-IF conversion, there is, as is well known, a splitting into two channels which are in quadrature with each other and which thus also require two analog-to-digital converters. Subsequent processing in a decoding device CE is implemented purely digitally and independently of the preceding stage E5.

The digitized signal following analog-to-digital converter ADE is now converted by a quadrature mixer M4 and decimation stages, not shown, such that the data rate of the resulting data stream corresponds to the symbol rate t_s or an integral multiple thereof. Quadrature mixer M4 is fed by an oscillator Os3 with a sine and cosine component of the down-mixed carrier frequency which also produce two mixing components at the output of mixer M4. To illustrate this, the data lines for these two components are shown as double lines in Figure 7. In the event the preceding circuit unit E5 is a zero-IF converter or low-IF converter, then two in-quadrature data paths in the low-frequency position are already present and quadrature mixer M4 is omitted.

The components in the two data lines involve digitized signal values which are, however, coupled to the transferred symbols. An electronic switch Sw1 now distributes these values synchronously at symbol clock t_s to two switch outputs 1, 2, thus supplying the inputs of a symbol detection device SD.

By means of switch Sw1, the signals from mixer M4 are alternately divided between the two inputs 1, 2 of symbol detection device SD, at the output of which the decided symbols can be tapped from the received signal. Based on the alternating division and subsequent solution of the linear equations for the received signals in linear combination h^{-1}, the preliminary estimated symbols A', B', or C', D' of each symbol pair Sy1, Sy2 are available at the combination's outputs. A decision element ET generates the decided symbols A", B", or C", D" therefrom which are converted by a following table TB into electronic data for symbols A, B, C, D for further processing. From the parallel available symbols A, B, or C, D of the symbol pairs, a switch Sw2 alternately controlled at symbol clock t_s regenerates the original data sequence D_0 with data A, B, C, D. This data stream can then be converted into the desired audio signal.

During decoding of the symbols, specifically, in the zero-IF or low-IF methods, the situation may occur in which the carrier is placed in an active frequency band during mixing. As a result, a large steady component is generated in the down-mixed signal, which component generally exceeds the operational ranges of the analog-to-digital converters. In the process of down-regulating the signal value, resolution is lost. It thus makes sense to chose another approach in which a simple control loop is used to

superimpose a sufficiently large direct component on the analog signal before digitization until the signal is more or less within the control range of the analog-to-digital converter(s).

The adaptation of the parameters in the linear combination h^{-1} is implemented by sending the signals of inputs 1, 2, and the two outputs of symbol decision element to one input each of arithmetic unit RE for comparison. In the steady-state condition, received symbols A', B', C', D', and decided symbols A", B", C", D" should be linked by the inverse transfer functions h_1^{-1}, h_2^{-2} in the linear combination h^{-1}, up to the point of unavoidable noise components, since the inverse transfer functions should of course exactly compensate the transmission paths. Deviations in linearity are determined by the equation systems in arithmetic unit RE and generate correction signals which are supplied by a control loop filter Fr to correction inputs of the linear combination h^{-1}.

For the purpose of conversion to the audio signal, however, additional information is required, such as the volume, tone, or balance which are a function of the specific location of the audio reproduction device. Additional control information relates to the location of the device within the three-dimensional sound system, that is, its address, the data compression method used, information on the applicable protection measures to secure data during transmission, and synchronization bits to detect the data package beginning and to synchronize symbol detection. This control information must be inaudibly superimposed on the actual audio signal, or transmitted in addition to this signal. The obvious advantageous approach for transmission here is the packet format which contains all the requisite control information and addresses in a header. The actual data component then contains the data for the audio signal, and optionally also the check bits or empty bits to fill out the individual data ranges.

Since the source data streams are sometimes already digitized, a sampling rate conversion or even recoding with a detour via an analog signal should be avoided. This however requires the transmission of such different sampling rates as 44.1 kHz or around 48 kHz, and integral multiples thereof. The selected data packet structure, usually called a frame, is advantageously 10 ms long. Following a header with synchronization bits and control parameters, two stereo blocks with 2 x 240 6-bit values each are transmitted at 48 kHz. At 44.1 kHz, three stereo blocks with 2 x 147 6-bit values each are transmitted.

At 44.1 kHz and lower sampling rates, the extraneous bits in the individual data blocks are filled with a predefined bit sequence.

Figure 8 is a schematic view of the above case showing the data formats for transmission of the audio data. Both data formats represent one data packet FD each of 10 ms length. The top data format is especially well suited for a source rate of 48 kHz, while the bottom format is well suited for a source rate of 44.1 kHz. The individual data blocks for the left and right audio channel L or R alternately follow the header H. In advantageous fashion, compression is oriented by pairs to these blocks such that the decompression (see arrow “decomp.” in Figure 8) can begin on the receiver side each time after reception of the first audio block pair L, R. In the top frame, this corresponds to a delay of about 5 ms, 3.3 ms for the lower frame. On the transmitter side, approximately the same delay value is added, with the result that the specification of lip synchronicity which requires a delay of less than 20 ms between video and sound can be met.